The set K_u consists of the index of the sounds used for the transmission of information. The column vector $h_u[n]$ is defined by

$$h_U^T = [h_{u_1}[n]\ h_{u_2}[n]\ \ldots\ h_{u_U}[n]] \qquad \text{with} \qquad \{u_1, u_2, \ldots, u_U\} = K_U\,. \tag{110}$$

$h_u[n]$ contains the pulse responses of the sounds used at the instant of time n. U is the number of the elements in the set K_u. The column vector $A[n]$ contains values that are applied to the sounds used.

$$A^T[n] = [A_{u_1}[n]\ A_{u_2}[n]\ \ldots\ A_{u_U}[n]] \qquad \{u_1, u_2, \ldots, u_U\} = K_U. \tag{111}$$

The x^{th} line of the matrix $C[n]$ is set up by the line vector $c_{u_x}^T[r]$,

$$C[r] = \begin{pmatrix} c_{u_1}^T[r] \\ c_{u_2}^T[r] \\ \vdots \\ c_{u_U}^T[r] \end{pmatrix} \qquad \text{for} \qquad r = 0, 1, \ldots, R-1 \,. \tag{112}$$

As already mentioned herein above, the signals to be obtained should be real value time domain signals. This includes that only the sounds 1, 2, ..., M/2-1 can carry any desired complex values, the sounds M-1, M-2, ..., M/2 + 1 must therefore transmit the conjugate complex values. The sounds 0 and M/2 are charged with zero. The sets K_u and K_I only contain the sounds below M/2 so that $K_u \leq \{1, 2, \ldots, M/2-1\}$ and K and $K_I \leq \{1, 2, \ldots, M/2-1\}$ has to be true. Charging of the sounds above M/2 with conjugate complex values is considered by the term CC, CC standing for conjugate complex. One sound can either be used for transmitting information or for compensation, it cannot be used for both though. The intersection of the sets K_u and K_I must be empty.

The equation (109) allows to specify the block diagram of the embodiment of the method according to the invention which is shown in Fig. 33. The upper portion of the block diagram consists of an IDFT 10 and is a customary DMT transmitter. The already mentioned influence arises from the weak spectral limitation of the base functions of the IDFT. The data vector $A[r]$ is applied to the sounds u, $u \in K_u$. In order to ensure a real value time domain signal, the conjugate complex data are applied to the sounds v, $v \in K^c_u = \{v \mid v = M - u,\ u \in K_u\}$.

To ensure simple computation on the side of the receiver, a Guard Interval or a cyclical prefix has to be introduced.

First, the case of a Guard Interval will be considered, but later on, the results obtained will be generalized to a system with a cyclical prefix. If the Guard Interval used has a length P, P zeros are introduced between two consecutive time domain blocks.

Each block 11, 12, 13 in Fig. 33 designated with $C[r]$, r = 0, 1, 2, ..., R-1 effectuates a multiplication of its input vectors by the matrix $C[r]$ in using the method according to the

invention. The collected results of these multiplications are applied to the compensation sounds. The input vectors applied to the blocks **C**[r] are the actual data vectors **A**[m] and delayed versions thereof, **A**[m-r], r = 1, 2, ..., R-1. At each time stage, the combined result is applied to the sounds i, $i \in K_I$. In order to ensure a real value time domain signal, the sounds i, $i \in K^c{}_I = \{i \mid i = M - i, i \in K_I\}$ must again be charged with the complex conjugate values.

Calculation of the weighting factors is carried out as follows. The matrices **C**[r], r = 0, 1, ..., R-1 have to be calculated in order to minimize the integral of the weighted power density spectrum. In order to be able to do this, an analytic expression of the power density spectrum has to be indicated. This is achieved by calculating the autocorrelation function $R_s[m]$ of the transmitted sequence $s_n[n]$ and by applying the Fourier transformed thereto, this yielding the power density spectrum $S_s(e^{j\theta})$.

As s[n] is a cyclostatic process, the time-dependent function $R_s[n, m]$ must first be calculated at the time n and the delay m

$$R_s[n,m] = \mathrm{E}\{s^*[n]s[n+m]\} \tag{113}$$

Upon substituting the equation (108) in (113) and some algebraic conversions, the equation (113) becomes